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# The Essentials Series

# The Business Value of SIP VoIP and Trunking

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## Article 1: VoIP in the Mainstream

Early man used pictographs on cave walls to express ideas. Since then, technologies have evolved to bridge the gap between humans and change how we interact. Communications have evolved as well. Today, we pick up the phone and call someone on the other side of the world. The delivery of our information, or ideas, is nearly instantaneous. Our culture has shaped us to expect things to happen very quickly, and we have become impatient about delays in delivering information in any form.

Over the past 10 years, Voice over IP (VoIP), once seen as a niche, has become a mainstay of converged services. What began as a quirky technique for communications between PCs on the Internet has grown into a sustainable mainstream technology that fully supports consumers, enterprise business, and carrier-grade operations. Today, VoIP provides the network foundation for the complete integration of voice, video, and data services with enterprise business applications. To appreciate the technologies of today, we need to consider the evolution that brought us here. The development of communications has been filled with wondrous advances in our ability to share information and ideas.

### The Evolution of Telephony

On March 10, 1876, Alexander Graham Bell made the first telephone call, and we were given a glimpse of a technology that would spread around the world and change how we communicate. The Public Switched Telephone Network (PSTN) has been growing ever since.

Some of the most significant events in the evolution of the telephone network weren't directly related to technology. When consumers were offered flat-rate pricing for local calling, usage of the telephone rose dramatically. In the mid-1960s, AT&T offered direct-dial long-distance service and for the first time users could call friends or family across the country without the intervention of a long distance operator. Again, this precipitated a dramatic rise in telephone calling patterns.

One of the key technologies that led to where we are today was the digitization of the telephone network. Telco central offices are connected by dedicated circuits called *trunks*. The Bell System led the telecommunications world for years and undertook a massive migration to convert these trunks from older analog circuits to a new digital technology. This migration was driven by the need to increase the density of network carrying capacity while striving to reduce network operating costs. T-1 circuits became a common trunking solution, providing 24 simultaneous voice paths between offices. This also allowed for switching or cross-connecting circuits carrying phone calls through multiple central offices more quickly and efficiently.

This T-1 methodology still exists today, but rather than time division multiplexed (TDM) circuits carrying 24 voice calls, many of these circuits now carry IP-based traffic. Digitizing the network trunking taught the industry a great deal about digitizing and packetizing voice calls. Although VoIP may be a new technology to many, the practice of digitizing and packetizing voice traffic has been in use for many years.

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## VoIP Collides with Telecommunications

As the lessons learned in digitizing voice traffic advanced, the idea of making a phone call over the Internet continued down a parallel path. Standards began to emerge and overlap. The telecommunications H.323 umbrella protocol standards for voice, video, and collaboration tools were widely adopted by the emerging VoIP community working on Internet technologies. Session Initiation Protocol (SIP) gained wide acceptance over 10 years, and today has become one of the most commonly used protocols.

Early fears that the Internet might reduce telephone network traffic have long since been allayed. Over time, we've seen that the Internet and the PSTN have not been able to replace the other. But the two networks continue to move closer together. Each adopts and incorporates technologies of the other that allow for seamless integration. Today, many callers on the PSTN are using traditional phones and never realize that their call is being carried by a VoIP service somewhere between the caller and called party.

### ***VoIP as a Mainstream Building Block of Voice Service***

For the telecommunications carriers, VoIP became a fundamental component of the voice service architecture. IP networks, although not providing the same inherent Quality of Service (QoS) level as a dedicated circuit on the PSTN, provide more resilience, bandwidth capacity, and versatility than the traditional TDM circuits did in the past.

#### **Circuit Switching**

Circuit switching is the legacy method used to establish a dedicated electrical (or optical) path between devices. This path or circuit is established for the duration of the telephone call and is dedicated to that call. These resources in the network cannot be used for other calls or by any other user until the call is completed and the resources are released and available.

Because networks cannot be designed to support every possible telephone call at the same time, the switches are designed as "blocking" switches. Thus, when all available resources are in use, callers will experience queuing delay or blockage until resources become available.

The delay through the network once a connection is made is minimal. Most of the telephone network has been designed to provide about 55 milliseconds of delay over the circuits that are established.

Because circuit switching guarantees a dedicated path that cannot be shared, it requires a significant engineering effort to locate, reserve, and connect the necessary resources through the network. This requirement drives up the cost of a connection and causes some delay in the setup process. As a result, circuit switching is more economical for connections of a longer duration such as a voice telephone call where parties may talk for 3 or 4 minutes. It works best when network utilization of the network is high, providing a usage level that keeps the resources busy but not overloaded.

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IP networks have enabled the carriers to reduce central office costs, incorporate broadband services such as DSL for Internet access, deliver video, and expand into new services. IP networking over packet switched networks delivers benefits to everyone using the network.

### **Packet (Store-and-Forward) Switching**

There are many ways to provide switching without the use of dedicated facilities. One example of a store-and-forward switching network is the subway system in New York. Passengers can travel between any of the subway stations along the route. The topology of this network is referred to as a “hub and spoke” topology. The subway has many switching points or nodes. To get from one location to another, users might have to transfer from one line to another at one of these nodes. At the hub nodes, passengers (the traffic) might have to wait in a buffer (be stored) until the next available train arrives so that they can move (be forwarded) to their destination. Just as passengers encounter delays in waiting for a train to arrive, and sometimes queuing delays when the trains are fully loaded, a store-and-forward type network provides service that has very different characteristics from a circuit switched network, which would be more similar to a New York taxi; dedicated to a passenger for the duration of the trip

In a data network, even the links between the switches are shared on demand. Switches perform routing calculations to determine which link to send the data onto, and then the data is placed in queue for that link. Resources are allocated on a first-come, first-serve basis, and there are no guarantees that the next leg of the path will be available upon arrival. Delays in queuing can cause data to sit in buffers. In a data network, this means that the delay through the network can be sporadic and unpredictable.

Because of this unpredictability, large blocks of information aren't well suited to this type of network. Large blocks of information have to be broken into smaller chunks in order to avoid degrading performance of the network. When we think of a 4-minute telephone call, we are really thinking of a very large block of data.

In a store-and-forward network, each block of data has to carry some form of addressing information that the switches can use to determine the location of the final destination. Without this, the information can never be delivered to the recipient.

Data applications are often described as being “bursty in nature,” meaning that there may be lapses or pauses between transmissions. Unlike a voice call, which is a real-time interaction between two people, a data connection is often an interaction between two computers without a person directly involved. Since store-and-forward, or packet, networks use statistical multiplexing, or first in, first out (FIFO) methods, this type of network is better suited to a bursty type of traffic, such as data.

Packet switching is the most common form of store-and-forward switching in use today, with routers being a perfect example of a store-and-forward switch. Packet switching breaks blocks of information into a pre-defined size or size range. This process of packetization creates some overhead, as each packet must have addressing information. Error checking can be performed on a per-packet basis, and if errors occur, only the corrupted packet needs to be retransmitted. This gains some efficiency in the network, as long as messages do not need to be repeated entirely if an error occurs.

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This advance from circuit switched voice to VoIP has provided huge benefits to enterprise business. Ten years ago, the term *convergence* referred to the idea of integrating voice and data services onto a single high-capacity circuit from the service provider to the business premise. Today, convergence has brought voice and data together on a single cabling infrastructure and network within the business, all based on IP. This integration of voice and data has set businesses who have embraced the change at the leading edge to now be able to integrate network services with enterprise business applications for improved efficiencies, reduced cost, and strategic business process improvement.

The evolution of convergence encompasses both Unified Communication and Communications-Enabled Business Processes. CEBP is an important way of thinking about the simplification and automation of communications as services on the network in conjunction with enterprise business applications. In a recent paper, Forrester Research defined CEBP as “business processes and applications tightly integrated with unified communications technologies to enable concurrent or consecutive communications among customers, suppliers, and employees within the context of business transactions.” Those organizations that have already migrated to VoIP, or are doing so today, are positioned to take the greatest advantage of complete integration.

## VoIP Penetration: Then to Now

VoIP technologies started deploying in the late 1990s. There were many early market efforts from a number of different carriers and solution providers. Some focused on consumers, and some on enterprise business. VoIP didn’t mature or achieve critical mass overnight.

Today, looking at quarter-by-quarter numbers from industry analysts, it appears at first glance that the market is slowing, yet overall penetration is on the rise. The incumbent, traditional telecommunications carriers are clearly losing landline customers at a steady rate. Consumers are migrating to VoIP solutions from cable companies and mobile services (both sectors clearly on the rise).

On the consumer side of the equation, according to iLocus, a Business-to-Business (B2B) market research firm that tracks the VoIP market, a recent bulletin noted the following:

In 1Q08, vendors shipped a total of about 7.9 million VoIP Subscriber Feature Server licenses for deployment in service provider networks generating \$144 million in revenue. The number of lines is down by 19% Q-o-Q. The 3Q07 and 4Q07 quarters however were unusually high growth quarters for VoIP Subscriber Feature Servers. If the 4Q07 seasonality in particular is normalized, there is a nice sequential growth.

iLocus does not track IP upgrades to TDM ports, which encompasses a huge portion of the enterprise market space. They focus on tracking Class 5 next generation network deployments, including:

- VoIP hosted telephony implementations
- New “greenfield” VoIP deployments
- Complete replacement of legacy switches with VoIP
- Extension of existing legacy networks with VoIP equipment in new geographic areas

Telegeography Research recently reported that as of the end of March 2008, 13.8 percent of US households, 27 percent of those with broadband, or 16.3 million consumer lines are using VoIP. When they look at VoIP provided by the cable companies, Telegeography reports the following numbers (descending):

Cable Provider	Number of Subscribers
Comcast	5.1 million
Time Warner	3.17 million
Cox Cable	2.46 million
Cablevision	1.68 million
Charter	1.08 Million

Telegeography’s report cites that growth in the cable companies’ subscriber base comes from the traditional old Regional Bell Operating Companies (RBOCs), who lost 17.3 million consumer residential phone lines during the period covered.

From a revenue perspective, Infonetics reported the following revenue projections:

Market Forecast	2008	2009	2010
IP Centrex	\$153M	\$200M	\$255M
IP Audio Conferencing	\$125M	\$165M	\$211M
Unified Messaging	\$27M	\$39M	\$54M
Generic SIP Application Server	\$34M	\$30M	\$22M

## The VoIP Evolution Simplified

The traditional circuit switched PSTN is a connection-oriented network. The connection is the call setup process that establishes the circuit between the parties on a phone call. Packet networks, particularly those based on IP and Internet technologies, can be either connection-oriented or connectionless. In a connectionless network, no setup is required. Each packet carries sufficient addressing or routing information to allow it to be passed from node to node through the network.

Unless quality mechanisms are put in place, there are no guarantees for QoS, but there are no dedicated network resources. The resources of the network can be shared and used efficiently. Connectionless networks also don’t inherently guarantee that packets will be delivered in the order they were transmitted. Packets might take different paths through the network and arrive at different times. Thus, the device at the recipient must have resources to store the packets until enough have arrived to reassemble the message for delivery. As you can see, packet networks provide a good technology for delivering short or bursty messages that don’t require the overhead of establishing a circuit connection during call setup.

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## Summary

VoIP leverages all the knowledge developed by the technology sector over the past century. The legacy telephone service providers understand traffic engineering, busy hour requirements, call blocking algorithms, and every nuance of delivering high-quality voice services. The IP or Internet industry learned about bundling the intelligence of addresses within packets, digitizing content, building QoS guarantees into a network the inherently has none, and building a massive scalable integrated service network. VoIP brings the best of all this mature technology together in a solid, reliable foundation that gives enterprise business the strongest possible architecture to build the foundation of the mission-critical business network.

## Article 2: The Transition of Trunking

This article will take a look at the evolution of telecommunications trunking technologies from traditional time division multiplexed (TDM) trunks and tie lines based on T-1 circuit infrastructure to IP-based trunking and built-in Session Initiation Protocol. SIP adoption has become a vital success factor on the road to unified communications and is critical to comprehensive integration with enterprise business applications to reach the Communications Enabled Business Processes (CEBP) of tomorrow. More than another standards-based protocol, SIP has become the *de facto* standard for unified communications that bring voice, video, and data together.

### Traditional Trunking Technology Evolution

In the evolution of traditional telephony, trunking technology dates back to very early in the maturation of the Public Switched Telephone Network (PSTN). In this article, we can't begin to address the complexity of the evolution of the PSTN, but the outline in Figure 1 highlights major milestones from early in the history of telecommunications to the present.

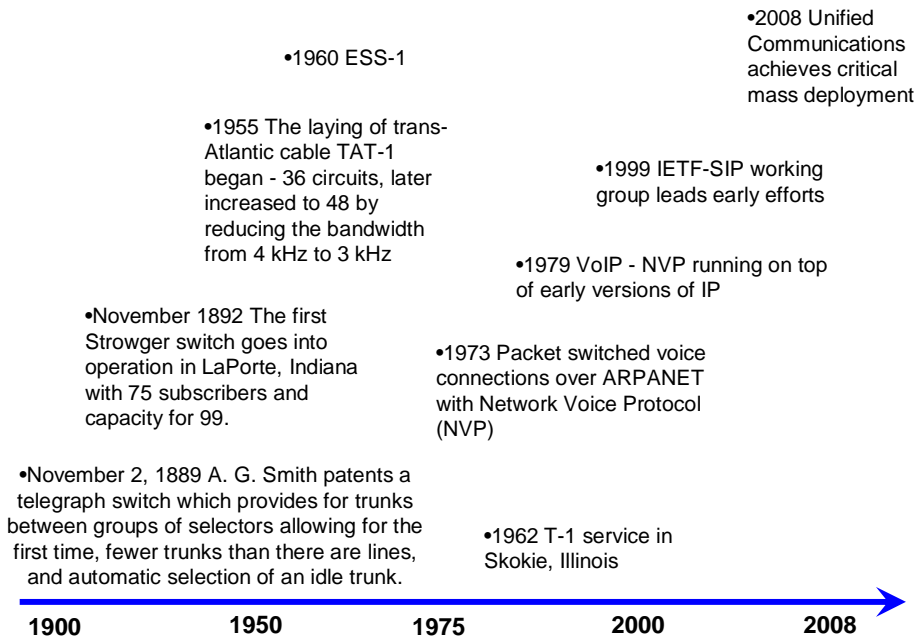



Figure 1: Telecommunications trunking evolution timeline.

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Trunking circuits have provided several functions through the years as telecommunications networking became more complex, adding features and services. Some trunk circuits were used for outgoing calls. Others handled the supervision of incoming trunks. Traffic Supervision Position System (TSPS) trunks were used to provide a connection to the telephone operator. TSPS, and other trunking technologies, were based on *analog* circuit technologies. TSPS was later replaced by Operator Service Position System (OSPS) to incorporate the features of the Class 5 Electric Switching System (5ESS) telco switch that was introduced in 1982.

For long-distance phone calls, trunks could be connected from one to another (called *tandem trunking*) to create voice paths that traversed the continent and circled the globe. Trunking circuits also provided tie lines between enterprise Private Branch Exchange (PBX) systems.

The biggest change to trunking technology before SIP was the digitization of the PSTN. After the first T-1 service in Skokie, IL went live in 1962, the Bell System—the dominant provider in North America—embarked on an aggressive effort to digitize the entire PSTN. This conversion took many years as *digital* circuits replaced their analog predecessors.

 For a detailed explanation of digital versus analog technologies, see *The Fundamentals of Packetized Voice*, Chapter 3 of *IP Telephony Demystified* (McGraw-Hill, ISBN 0071406700).

## The Role of Multi-Protocol Label Switching in the Network

Multi-Protocol Label Switching (MPLS) is a widely used mechanism for delivering Quality of Service (QoS) in packet switched networks. In IP networks, MPLS eliminates the normal hop-by-hop routing in IP from the equation. MPLS adds a *tag* to each packet. This tag shortcuts the delivery path by sending packets to the best available route for a given traffic type. MPLS has been widely adopted in both enterprise and service provider networks. As business networks integrate Voice over IP (VoIP) service, these QoS enhancements are frequently needed to support the growing volume of VoIP and now video traffic.

MPLS has often been referred to as a bypass or “shim” protocol. The insertion of a tag into the packet stream adds minimal overhead that is easily offset by the enhancements of service. MPLS is often referred to as a *Layer 2½ protocol* because it straddles Layers 2 and 3 of the OSI Model.

 See <http://computer.howstuffworks.com/osi.htm> for a simple, online explanation of the OSI Model.

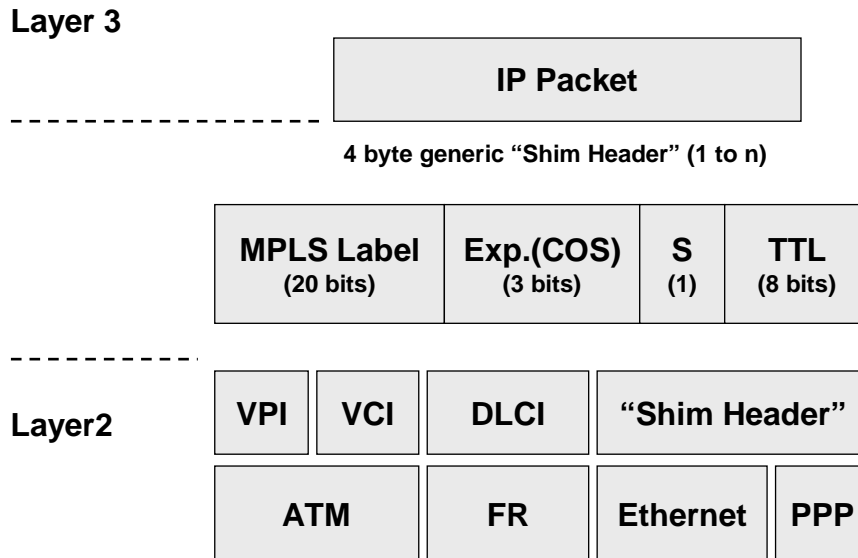
### Using MPLS for QoS

Implementing VoIP makes many enterprises reassess network capabilities. The readiness assessment performed in the planning stage of VoIP deployment typically presents findings that require this evolution. MPLS can further introduce fundamental changes to the way these IP networks operate. VoIP service brings a change in thinking about the expectations for the network. MPLS allows IP networks to mimic some of the behavioral characteristics of a circuit-switched network like the PSTN.

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## MPLS Structure

MPLS works by inserting an MPLS shim header, or *tag*, into the beginning of each packet. This shim header contains one or more *labels* and is often called a label stack.



**Figure 2: MPLS packet labels and encapsulation.**

The MPLS label stack entry contains four fields:

- A label value
- Class of Service (COS)
- A flag to indicate whether this label is the last label in a stack
- A time to live (TTL) value

MPLS performs a label lookup/switch instead of a lookup into an IP routing table. Label switching can be performed within the switching fabric of the hardware, so it's faster than typical routing.

### QoS vs. QoE

When implementing VoIP services, it's prudent to separate thinking about QoS and Quality of Experience (QoE). QoS parameters are typically systems-based metrics, including delay, jitter, and packet loss. QoE metrics are directly related to services such as VoIP. VoIP QoE metrics address areas such as call completion rate, call setup time, and audio quality—the factors that a person on the phone notices.

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## **Traffic Aggregation for Similar Traffic Classes**

The COS field designates the assignments for different classes of service. The common and simplest view is to assign three service classes for user traffic and a fourth for management traffic. Typical MPLS classes of service are

- Real-time traffic—Voice and interactive video, for example, are almost always assigned the highest priority. This assignment not only guarantees adequate bandwidth but also supports providing the delay, packet loss, and jitter characteristics needed for delivery of real-time traffic such as VoIP and video.
- Mission-critical data traffic—Traffic such as that from a legacy mainframe is usually placed in a call by itself that guarantees delivery. Delivery timing is often a more stringent requirement for this type of data than other factors.
- All remaining traffic—The leftover traffic is usually aggregated into a best efforts service class that simply mirrors how IP normally handles traffic delivery.
- Management traffic—This traffic is commonly aggregated into a management class by itself because it requires assurances that it can still be delivered even during periods of heavy network congestion.

## **MPLS Virtual Private Networks for Privacy and Aggregation by Business Class**

Using these techniques, not only can similar traffic types be aggregated into the same class of service but it can also be separated into what are typically called Virtual Routing Forwarding (VRF) tables. This aggregation creates a forwarding equivalency class (FEC) for switching throughout the network. These VRFs are used for separation of traffic for privacy or for traffic engineering.

This approach creates a virtual private network (VPN) environment. It all takes place at the MPLS header level, so traffic destined for one destination—whether a customer or a business division in an enterprise—can never be seen by another. Service providers commonly use MPLS-based VPN aggregation; they provide QoS markings within each VPN and at the egress points to other networks.

Using MPLS encapsulation techniques, layers of MPLS labels are built, enabling hierarchical switching of MPLS packets. This approach enables a carrier to deliver the privacy of a dedicated network coupled with the QoS guarantees required to support real-time traffic such as VoIP.

### MPLS Resources

MPLS Resource Center at <http://www.mplsrc.com/index.shtml>

MPLS MFA Forum at <http://www.mplsforum.org/>

IETF MPLS Working Group at <http://www.ietf.org/html.charters/mpls-charter.html>

IETF RFC 3031 at <http://www.ietf.org/rfc/rfc3031.txt>

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MPLS can't be compared directly to IP. It's a complimentary protocol. MPLS works in conjunction with IP and IP's interior gateway (IGP) routing protocols. MPLS introduces the capability of traffic engineering to the IP network.

MPLS and IP's routing protocols are the glue that bind a network together and ensure it works as designed. Border Gateway Protocol (BGP) enables the concept of peering between organizations for passing or sharing traffic routes. This BGP peering conceptually provides a nice way of thinking about SIP trunking. Just as BGP is the glue that binds the Internet together as a *network of networks*, SIP is the glue that binds unified communications systems together for VoIP and other communications services.

## Introduction to SIP

Intense work on SIP really began in earnest with the Internet Engineering Task Force (IETF) in 1999. It was one of many efforts and has been led by the IETF-SIP working group. Their charter states that SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users. Such sessions include voice, video, chat, interactive games, and virtual reality. This group has worked long and hard to help SIP mature. What began as a series of proposed drafts and standards, including numerous extensions, has become a foundation for VoIP and unified communications.


The basic model and architecture defined for SIP sets out some specific characteristics:

- Wherever possible, SIP services and features are provided end-to-end
- Extensions and new features must be generally applicable; they cannot apply only to some specific set of session types
- Simplicity is key
- Existing IP protocols and architectures are re-used and integrated tightly

SIP uses an addressing structure similar to email addresses. Users may log in anywhere and be dynamically assigned an IP address, so there has to be a way to resolve some of the common conventions in the active and current IP address.

SIP is text based, so the addresses, which are SIP URLs or URIs (Uniform Resource Locaters or Indicators), can be imbedded in email messages or Web pages. Additionally, as SIP is a text protocol, SIP URLs and URIs are network-neutral. Thus, a URL might point to an email-like address, using SIP, an H.323 address or even a telephone number on the PSTN.

SIP operates independently of the IP network layer. It requires only unreliable packet delivery and provides its own reliability mechanism. Although it's widely used in IP networks today (usually over UDP to avoid the overhead of TCP), SIP can run over IPX, Frame Relay, ATM, AAL5, or X.25 with no changes.

 There are hundreds of references on the Internet for readers who want to learn more about SIP. For example, there is an excellent set of resources maintained by Columbia University at <http://www.cs.columbia.edu/sip/>. Wikipedia also has a good starting point article at [http://en.wikipedia.org/wiki/Session\\_Initiation\\_Protocol](http://en.wikipedia.org/wiki/Session_Initiation_Protocol).

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## **Evolution of SIP Beyond Phone Calls to Trunking**

Why does trunking matter to the enterprise? One of the most obvious reasons is that SIP trunks reduce telecommunications expenses. For a recent explanation, Scott Lowe reviewed the situation at Westminster College on the Tech Republic Web site (<http://blogs.techrepublic.com.com/tech-manager/?p=501>). First, let's look at why the interest in change:

*Between T1 costs, a "billing charge" and local and long distance usage, Westminster College spends quite a bit of money on communications costs each month. The billing charge is a several-hundred-dollars charge that we incur for the privilege of receiving an itemized statement at the end of each month. We're a relatively small place, so we have just a single T1, which is dedicated for outgoing calls. For incoming calls, we have a pair of PRI circuits in place and, because of rather strict contract termination terms, will stay in place for the next couple of years. Our outgoing T-1, however, is up for grabs.*

Their primary driver was expense. Cost is a big driver for change in any enterprise, and VoIP as a cost-reduction tool has often provided the impetus for change. For many organizations, that is the first step toward unified communications.

As Lowe describes the situation, SIP trunks to replace the existing T-1 resulted in a significant cost savings, and the inclusion of all local and long-distance calling within the US. According to their calculations, a move to SIP trunks would cut usage charges by 95 percent.

An added incentive for many organizations is that SIP trunks can be delivered over existing Internet connections. Thus, when they aren't in use, the bandwidth is still available for other business activity. When you compare this broad use of SIP trunks against existing T-1 lines that mostly sit idle, it is a measurably more efficient use of resources.

A SIP trunk can be connected to any IP-PBX that supports SIP. Most manufacturers of IP-PBXs support SIP today. At Westminster College, they're looking at the Avaya IP Office PBX as a solution.

If you use existing Internet connections, keep QoS and QoE issues in mind. If you have enough bandwidth and your Internet provider has appropriate peering agreements, quality is likely to be quite good. But remember that SIP runs over IP. Your solution, if not properly designed, might be more prone to dropped calls or echo than with legacy voice circuits to a traditional PBX.

SIP has been widely adopted by many of the fixed-line carriers. They are all migrating their core networks to VoIP. SIP is widely used in the gateways for the connections between these carriers.

SIP itself has evolved. Its early focus was on the idea of a peer-to-peer communications protocol. Today, SIP has become the centerpiece of the family of protocols delivering our next-generation networks (NGNs). SIP's biggest near-term growth will be in interworking, or SIP trunking, to connect IP-PBXs and communication systems to IP service providers.

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## Summary

SIP will continue to extend the communications market. IP-based PBXs are increasing in use as traditional TDM systems reach the end of usable life, with applications now taking center-stage. The adoption rate of VoIP gateways, handsets, and other components is on the rise.

As enterprises adopt unified communications strategies as a business tool that integrates voice, video, and data services onto a single, converged network, SIP is truly the unifying protocol that brings all communications together. SIP sets the foundation for reduced cost, improved efficiency, and the next generation of CEBP.

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## Article 3: The Convergence of Trunking with the Platform

As voice traffic moves to Voice over IP (VoIP) and legacy time division multiplexed (TDM) trunking technologies shift more to Session Initiation Protocol (SIP), the two come together to deliver an environment for enterprise business that approaches nirvana. SIP connections, or *trunks*, were originally used internally by VoIP providers, then quickly matured into a service offered by Internet Telephony Service Providers (ITSPs) that enabled enterprise connections from a corporate PBX system to the Public Switched Telephone Network (PSTN). These SIP connections simplify administration because the SIP trunk integrates voice and data onto a single circuit connection, eliminating legacy TDM T-1 circuits. With the integration of SIP, features such as Caller ID and other call-related information is easy to transmit.

Growth of VoIP in the consumer market space continues to fuel broad industry growth that feeds enterprise adoption as well. According to an In-Stat report, the consumer VoIP market will reach 38 percent of broadband households worldwide by 2011. That means the consumer VoIP revenue stream will grow from \$15 billion to nearly \$44 billion. The enterprise adoption rate parallels consumer numbers today but has recently begun to rise more sharply.

### Putting Together the Convergence Pieces

For enterprise business, convergence has been a process that has unfolded slowly. What began as a move to consolidate circuit and administrative costs has become an integration tool that delivers increased productivity, improved business process workflows, and opportunities to deliver services in new and profitable ways.

#### ***VoIP as a Building Block in the Foundation***

As organizations have deployed VoIP services, one common thread has been noted consistently: VoIP is a building block element. VoIP provides the foundation for other emerging and next generation services such as unified communications (UC). For many enterprises, SIP trunking wasn't a consideration when the VoIP strategy was devised. Like the Information Technology Information Library (ITIL) framework that many businesses have adopted, there is a maturity model mentality that takes hold. As enterprises adopt VoIP, they realize the benefits of integration and convergence. New technologies lead to new thinking. This new thinking is driven by the new foundation of voice services technologies that support the core business drivers.

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For enterprises still behind the adoption curve, there is a compelling impetus to change in order to survive in the intensely competitive market. In a recent interview, Charles Larmond, a leading telecommunications industry consultant, was asked why sales of VoIP and SIP solutions are increasing so rapidly. His response raises a number of facets identifying the importance of aggressive adoption.

*What many businesses have right now is a clogged financial drain with pressure building up behind it. Companies are getting backed up with bills and are just starting to look for new ways to free up the cash flow to keep their operations going. At some point, they have to take action. A lot of companies are cutting back on inventory and salaries, which can hurt their ability to bring in new revenues. But many haven't looked at their phone bills for years and years. And a lot has changed since then. It is possible for a company to save 40 or even 60% on their phone bills by switching from traditional telephone service to VoIP. While that may not be enough to unclog the drain, it can be enough to keep a business alive long enough to survive the current economic conditions. Smart Telecom agents are starting to go out and offer to overhaul the telephone systems and contracts for these companies. I haven't been this busy in years.*

Inventories of manufactured goods, cash flow, salaries, and the like are vital business expenses. Larmond addresses the problem from a business climate perspective, but he really seems to take a survivalist view. Although a relatively sour economy encourages many companies to take a look at cheaper telecom solutions, forward-looking companies are highly motivated to find new business workflow efficiencies that can be delivered only through Communications Enhanced Business Processes (CEBP), part of Unified Communications.

## **Business and Competitive Differentiators**

In today's environment, businesses are aggressively pursuing measurable results from their investments. CEBP holds the promise of measurable business value by connecting the people, information, and workflow processes that support business objectives. This integration can be achieved only by organizations that adopt and leverage the benefits of VoIP services coupled with SIP trunking technologies.

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The following points from the iLocus Global VoIP Market 2008 Report highlight key indicators of the shift in momentum as business enterprises focus on change:

- *SIP trunking has gained significant traction over the last 18 months. As of April 2008, there were over 5 million end users connected to SIP trunks. SIP trunking service revenues are forecast to touch \$180 million during 2008 worldwide. By 2012, it is expected to be a \$5 billion / year market.*
- *Most telcos and switch vendors are working to get their Voice 2.0 (Communications Aware Mashups) developer programs in place. On one hand, the availability of high-level APIs such as the Web Services APIs opens application development to hundreds of thousands of Web developers. On the other, those developers who invest substantial amounts of money into their applications prefer to go direct to the end user, utilizing Open Source telephony platforms.*
- *Estimated 1.18 trillion minutes of VoIP traffic was carried by service providers worldwide in 2007. That represents a 35% increase over 2006 VoIP traffic. Of these minutes, 286.3 billion was local call volume, 815.2 billion was national long distance (NLD) call volume, and 81.4 billion was the international long distance (ILD) call volume.*

The aforementioned iLocus report coupled with another industry report from the combination of Wainhouse Research and In-Stat in late 2007 provides additional measurable data that points directly to enterprise adoption:

- *In the enterprise segment, an estimated 11.4 million desktop IP phones were shipped in 2007, thereby generating revenues of about \$1.13 billion worldwide (iLocus)*
- *Fifty-three percent of IT executives who participated in the Nemertes benchmark entitled Advanced Communications Services 2008, reported their organization was already using, or planed to use SIP trunking services in the next 1-3 years, 26% noting that they were planning for deployment in the 2008-2009 timeframe.*
- *A recent study by In-Stat and Wainhouse Research suggests the entire unified communications products and services market will be \$22.6 billion in 2007, growing to \$48.7 billion in 2012.*

When we take these numbers from the industry overall, there is a clear and compelling message for every business enterprise—VoIP and SIP are not speculative technologies of the future. They are the foundation of the business network of the present. They're the critical enablers for the next generation of communications and the current generation of successful business processes. They are the tools that empower business process redesign.

The coming widespread adoption of CEBP as part of UC will couple communications technologies with business processes and applications for improved communications channels between customers, suppliers, and employees. The advantages of this combination may include business techniques, best practices, administrative and management tools, and increased productivity. Convergence of services through VoIP and SIP is an invaluable tool in the competitive enterprise environment. It highlights focal areas in redesigning business processes to make the enterprise more effective and competitive.

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## IP-Based SIP Trunking as a Business Enabler

Many companies have already discovered the cost and productivity benefits of adopting advanced communication services ranging from MPLS and Ethernet to VoIP and SIP trunking. These tools enable organizations to integrate voice, data, video, imaging, and enterprise applications on a single enterprise network. Implementing these advanced services, IT teams do far more than enable effective business communications. They also deliver the tools to better serve customers, generate increased sales, and improve productivity.

### ***The Geographically-Dispersed Enterprise***

These capabilities are also reaching out across wireless networks to both remote and truly mobile workers. Unified communications environments are reaching across other communications modalities, requiring IT decision makers to fully understand the implications of IT planning and investment on supporting core business drivers and initiatives.

The rapid expansion of WiFi, existing carrier broadband solutions, and the services ahead such as 3G voice/data and WiMax set the stage for a business environment that is free from the constraints of a single location. Unified communications technologies built on VoIP and SIP enable the always-on, always-connected, transparent enterprise. This new enterprise raises the bar on cost efficiency, competitiveness, and profitability.

### ***Extranet Partners***

In the same way IP-based networking has become the standard for business interconnectivity, SIP trunking has become the *de facto* standard for voice communications. Person-to-person communications will always be a cornerstone of business. Whether the enterprise is oriented towards manufacturing of products or delivers services, the old adage that “people buy from people” remains true. Customer service and support relies on person-to person communications, but so do many aspects of supply chain management and partner relationships.

For the enterprise that has moved upward in the unified communications maturity model, SIP trunking as a partnering tool is commonly expected and required of trusted and valued business partners. Just as IP enabled the extranets, SIP enhances the extranet and provides a tool to strengthen the business work processes.

### ***Service Providers and the Evolution***

For service providers that deliver enterprise application software solutions, there are new expectations coming from enterprise businesses as a result of unified communications advances. Although an off-the-shelf business application may work for some businesses, most require tighter integration between network voice or data services and their workflows.

Strategic value as part of the value proposition a service provider brings is a competitive differentiator. Service providers that understand and appreciate the value of unified communications and their impact on business workflows and processes is demonstrated by how tightly integrated SIP solutions are packaged.

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## Why VoIP/SIP Matter

It can't be reiterated enough that VoIP and SIP are the foundation of unified communications. Just as IP routing protocols bind the Internet and make it work, VoIP and SIP are the glue that binds business communications. The value of legacy PBXs and TDM trunking is past. Today, those technologies are a business liability rather than an asset. Today, businesses require a solid unified communications framework to survive and compete. That framework is built on VoIP and SIP for call processing, trunking, and all communications services.

## Position for the Future

The move to the IP-based enterprise has been underway for several years. Unified communications are gaining momentum as businesses recognize the inherent value of integration, or convergence. The fully converged network enables immediate cost reductions in both staffing and maintenance of hardware and software. Implementing new services today requires measurably reduced capital investments than traditional systems, which simply no longer meet business needs.

IP's virtualization capability has resulted in significant cost savings. It has also eliminated geographic limitations. Workers can easily sit at a computer anywhere on the planet. Unified communications technologies virtually connect them to all enterprise voice and data services seamlessly and transparently.

Beyond that, the fully converged enterprise is positioned to implement new cutting-edge applications and services that offer significant improvement in business productivity, efficiency, and customer satisfaction.

## Maximizing Value

It's widely acknowledged that leveraging existing technologies remains one of the greatest challenges any organization faces in the evolution to unified communications. Businesses must understand and embrace new features and functionality. At the same time, most enterprise cannot abandon old applications upon which the company was built. These tools have driven success and aren't easily replaced.

For many organizations, the change is a migration process, integrating VoIP, SIP, and unified communications strategies site-by-site, group-by-group, or application-by-application. The fully converged IP network runs on a standards-based SIP infrastructure, so this migration can take place in a seamless, phased manner. This approach brings new value each step of the way.

The good news is that even with a small deployment, the benefits of SIP can be leveraged quickly. The evolution to a fully converged, unified communications solution is not an all-or-nothing proposition. There is no requirement to migrate the entire enterprise at one time. It's important to design a strategy around business requirements. As an open standards protocol, SIP promotes interaction and involvement between individuals and companies in shaping how IP communications can grow and evolve.

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## **Competitive Edge Today = Competitive Differentiation Tomorrow**

The future of CEBP on the horizon is a powerful and formidable competitive tool. It's the next evolutionary step of unified communications. Enterprises that embrace the fully converged network methods of unified communications will be the ones that are quicker to streamline business processes, adopt new workflows, and adapt to a changing landscape. Unified communications, encompassing CEBP, gives us a glimpse into an era of competition wherein the quick will get quicker. The nimble will become more nimble. The gap between market leaders and losers will quickly widen in the next few years. We can see a window in the time ahead when there will be spectacular leaps forward as innovative companies seize leadership positions in their respective markets. These will be the companies that built a solid unified communications framework and are prepared.

### **Summary: Don't Fail to Plan. Set the Stage.**

An explosion of unified communications solutions is underway. Smart business enterprises are moving quickly in an effort to stay competitive and win in their respective market segments. Voice services have been too oriented toward VoIP alone in the past few years. Now is the time to recognize that voice is one of many communications services of the network. Conferencing and collaboration tools are rapidly becoming crucial factors in unified communications.

The unified communications landscape is filled with opportunity across the spectrum of enterprise business applications. VoIP provided an entry point but not the end objective. SIP delivers a fundamental cornerstone that must be deployed to achieve a fully converged network. This fully converged network provides the first layer. Unified communications is the foundation for the future of CEBP and delivers the promise of the always-on, always-connected, always-growing, successful enterprise.